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EXAMINER	
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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No.	Applicant(s)	
	10/533,612	ASADA ET AL.	
	Examiner	**Art Unit**	
	Joseph Saunders	2614	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE *3* MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *11 December 2009*.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-20* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-20* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on *14 March 2008* is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☒ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☒ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☒ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____.

5) ☐ Notice of Informal Patent Application

6) ☐ Other: _____.

DETAILED ACTION

Continued Examination Under 37 CFR 1.114

1. A request for continued examination under 37 CFR 1.114, including the fee set forth in 37 CFR 1.17(e), was filed in this application after final rejection. Since this application is eligible for continued examination under 37 CFR 1.114, and the fee set forth in 37 CFR 1.17(e) has been timely paid, the finality of the previous Office action has been withdrawn pursuant to 37 CFR 1.114. Applicant's submission filed on December 11, 2009 has been entered. Claims 1 – 20 are currently pending and considered below.

Claim Rejections - 35 USC § 112

2. The following is a quotation of the first paragraph of 35 U.S.C. 112:

> The specification shall contain a written description of the invention, and of the manner and process of making and using it, in such full, clear, concise, and exact terms as to enable any person skilled in the art to which it pertains, or with which it is most nearly connected, to make and use the same and shall set forth the best mode contemplated by the inventor of carrying out his invention.

3. Claims 1 – 20 are rejected under 35 U.S.C. 112, first paragraph, as failing to comply with the written description requirement. The claim(s) contains subject matter which was not described in the specification in such a way as to reasonably convey to one skilled in the relevant art that the inventor(s), at the time the application was filed, had possession of the claimed invention. Applicant states on page 9 of the Remarks that support for the amendment is found in paragraph [0086], however while paragraph [0086] has support for adjusting a variable high pass filter it does not state "the audio signal at the second point in the sound field is <u>suppressed</u> relative to the frequency

response to the audio signal at the first point in the sound field". Therefore, since support is lacking for "suppressing" appropriate clarification and correction is required.

Claim Rejections - 35 USC § 103

4. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

5. Claims 1 – 4 and 10 – 13 are rejected under 35 U.S.C. 103(a) as being unpatentable over Bienek et al. (WO 02/078388 A2), hereinafter <u>Bienek</u>, in view of North (US 6,801,631), hereinafter <u>North</u>, and in further view of Kunugi et al. (US 4,868,878), hereinafter <u>Kunugi</u>.

Claim 1: <u>Bienek</u> discloses an audio signal processing method (method and apparatus to create a sound field) comprising the steps of: supplying an audio signal (input signal 101) to each of a plurality of digital filters (means 1506 includes signal delay means 1508, amplitude control means 1510, and adjustable digital filter 1512); respectively supplying outputs from the plurality of digital filters to a plurality of speakers arranged in a speaker array to form a sound field (Description of Figure 6, Pages 18 – 19); setting a predetermined delay time in each of the plurality of digital filters so that transmission delay times with which the audio signal arrives at a first point in the sound field via each of the plurality of digital filters and each of the plurality of speakers will coincide with

each other (Third Sound Field, Pages 21 – 22 and Figure 7C and Figure 8); and adjusting at least one amplitude characteristic of the plurality of digital filters ("The amplitude control means (ACM) is conveniently implemented as digital amplitude control means for the purposes of gross beam shape modification," Page 12 Line 25 – Page 13 Line 6).

Bienek does not explicitly state the amplitude control means is adjusted such that the frequency response to the audio signal at a second point in the sound field is lower than the frequency response to the audio signal at the first point in the sound field. Bienek does explain, "The amplitude control means is preferably arranged to apply differing amplitude control to each signal output from the Distributor so as to counteract for the fact that the DPAA is of finite size by using a window function," Page 12 Line 25 – Page 13 Line 6 and Figures 11A – 11D, "The window function reduces the effects of "side lobes" at the expense of power. The type of window function used is chosen dependent on the qualities required of the resultant beam. Thus, if beam directivity is important, a window function as is shown in Figure 11A should be used. If less directivity is important, a window function as is shown in Figure 11D can be used," Page 26 Lines 20 – 25, "Thus, in general, output signals destined for SETs near the centre of the array will not be significantly affected but those near to the perimeter of the array will be attenuated according to how near to the edge of the array they are," Page 12 Line 25 – Page 13 Line 6.

Again, while not explicitly stated by Bienek, North discloses a similar speaker system with multiple transducers positioned in a plane for optimum acoustic radiation

pattern and illustrates in Figures 20 – 23 that the frequency response to the audio signal at a second point in the sound field (off-axis) is lower than the frequency response to the audio signal at the first point in the sound field (on-axis). North explains "conventional arrays direct a portion of their sound towards the listener (main lobe) and a portion of their sound towards the sides (side lobes). When placed in a room, the sound waves in the side lobes reflect from surfaces such as walls, the ceiling, and the floor (room reflections). The reflected sound waves interact either constructively or destructively, depending on the delay time and frequency, with the direct sound waves. The listener is presented with a sound field that is a combination of the direct sound from the main lobe and the reflected and delayed sound from the side lobes. Although the speaker system may have exhibited a flat frequency response in an anechoic chamber, the frequency response in the room at the listener is anything but flat, with pronounced variations in the response. The result is the listener hears a severely distorted sound field that bears little resemblance to the original event," Column 2 Lines 22 – 52. North further illustrates on-axis and off-axis frequency response "waterfall" plots for different speaker arrays (Figures 20 – 23). North explains, "A waterfall plot illustrates the sound dispersion characteristics of a speaker system by cascading the multiple frequency response curves of a speaker system taken at multiple angles. The waterfall plots of FIGS. 20 through 23 represent 10 frequency response measurements taken in 10-degree increments from 0 degrees (on-axis) to 90 degrees off-axis. The measurement curves cascade diagonally on top of one another, starting with the 0-degree on-axis curve (the rearmost curve) and ending with the 90-degree off-axis curve

(the curve at the front), Column 6 Lines 34 – 44. North further explains, “with less energy directed towards the room walls, floor, and ceiling, the intensity of the room reflections decreases. This decreases the level of sounds that echoes within the room. A decreased level of room reflections can also result in a flatter frequency response in a room as the attenuated reflected waves interfere less with the sound waves in the main lobe,” Column 7 Lines 32 – 45.

Thus in all cases, North illustrates (Figures 20 – 23) that the frequency response to the audio signal at a second point in the sound field (off-axis) is lower than the frequency response to the audio signal at the first point in the sound field (on-axis), especially Figure 22, where North illustrates that a significant reduction to off-axis sound energy due to increased directivity results “in a flatter frequency response in a room as the attenuated reflected waves interfere less with the sound waves in the main lobe,” Column 7 Lines 32 – 45.

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention given the teachings of North to utilize the reduction in the effects of "side lobes" by using the amplitude control means by applying a window function to the audio signal as disclosed by Bienek to achieve a frequency response to the audio signal at a second point in the sound field (off-axis) that is lower than the frequency response to the audio signal at the first point in the sound field (on-axis) since it is beneficial to have a significant reduction to off-axis sound energy due to increased directivity given it results “in a flatter frequency response in a room as the attenuated reflected waves interfere less with the sound waves in the main lobe,” North Column 7 Lines 32 – 45.

Bienek and North further do not disclose adjusting a variable high pass filter to filter the audio signal such that the frequency response to the audio signal at the second point in the sound field is suppressed relative to the frequency response to the audio signal at the first point in the sound field. However, in line with the teachings of Bienek and North of "reducing the effects of "side lobes"," Bienek Page 26 Lines 20 – 25, and having "a flatter frequency response in a room as the attenuated reflected waves interfere less with the sound waves in the main lobe," North Column 7 Lines 32 – 45, Kunugi teaches another method that does more than just reduce "side lobes" it eliminates the unintended sound waves at the listening point in the sound field while compensating for sound waves of insufficient low frequency level. Kunugi explains, "As is apparent from the above description, in the sound field correcting system according to this embodiment of the invention, a signal obtained by varying the frequency characteristic of the original signal so that the reflected sound waves are eliminated at the listening point is added to the original signal to obtain the loudspeaker driving signal. Therefore, the frequency characteristic at the listening point can be made essentially flat. Furthermore, the system is simple both in the adjustment required and in its construction, and it can be manufactured at a relatively low cost," Kunugi Column 7 Lines 35 – 45. Kunugi further improves upon the first embodiment and explains, "As is apparent from the above description, in accordance with the second embodiment of the invention, the reflected sound wave is cancelled by controlling the delay and the level of the original signal, and furthermore the HPF having the cutoff frequency f_0 is inserted in the original signals's delay and level controlling system to boost the low frequency level.

In this embodiment, the cutoff frequency f_0 of the HPF 2-9 is determined according to the sound field characteristic established by the direct sound L.sub.1 and the reflected sound path L_2. However, in practice, the HPF 2-9 should be implemented with a variable resistor R_1, as shown in FIG. 11, so that the cutoff frequency f_0 can be selected as required. In FIG. 11, C_1 designates a capacitor. According to the second embodiment of the invention, the HPF is used to change the frequency characteristic of the signal which is added to the original signal, and therefore sound waves at the listening point are remarkably improved in low frequency level. That is, sound waves of insufficient low frequency level are compensated for, with the result that the low frequency reproduction capability is substantially increased (Column 8 Lines 33 – 58).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the circuit of Bienek and North, to eliminate the unintended sound components responsible for excess reverberation, poor channel separation, and irregular frequency characteristics as taught by Kunugi (Column 3 Lines 16 – 31) and further include a variable high pass filter as taught by Kunugi, allowing for the low frequency response level of the audio signal at the listening point to be boosted relative to the reflected waves at the second point which are suppressed through cancellation, thus improving the frequency response at the listening point of Bienek and North.

Claim 2: Bienek, North, and Kunugi disclose the audio signal processing method according to claim 1, wherein a sound wave from the speaker array is caused to reach

at least one of the first and second points after it is reflected by a wall surface (Bienek Figure 8).

Claim 3: Bienek, North, and Kunugi disclose the audio signal processing method according to claim 1, wherein when forming the first and second points in the sound field, a filter factor of each of the plurality of digital filters is determined by calculation and set for each of the plurality of digital filters (Bienek Third Sound Field, Pages 21 – 22 and Figure 7C).

Claim 4: Bienek, North, and Kunugi disclose the audio signal processing method according to claim 1, wherein when forming the first and second points in the sound field, a filter factor of each of the plurality of digital filters is read from a data base and set for each of the plurality of digital filters (stored sets of delays (for the DDGs) and filter coefficients (for the ADFS) can be recalled, Bienek Page 14 Lines 26 and 27).

Claim 10: Bienek discloses an audio signal processor (method and apparatus to create a sound field) comprising a plurality of digital filters (means 1506 includes signal delay means 1508, amplitude control means 1510, and adjustable digital filter 1512) each supplied with an audio signal (input signal 101), wherein each of the plurality of digital filters supplies an output signal to each of a plurality of speakers arranged in a speaker array to form a sound field (Description of Figure 6, Pages 18 – 19); each of the plurality of digital filters has a predetermined delay time so that transmission delay times with

which the audio signal arrives at a first point in the sound field via each of the plurality of digital filters and each of the plurality of speakers will coincide with each other (Third Sound Field, Pages 21 – 22 and Figure 7C and Figure 8); and adjusting at least one amplitude characteristic of the plurality of digital filters ("The amplitude control means (ACM) is conveniently implemented as digital amplitude control means for the purposes of gross beam shape modification," Page 12 Line 25 –Page 13 Line 6).

Bienek does not explicitly state the amplitude control means is adjusted such that the frequency response to the audio signal at a second point in the sound field is lower than the frequency response to the audio signal at the first point in the sound field. Bienek does explain, "The amplitude control means is preferably arranged to apply differing amplitude control to each signal output from the Distributor so as to counteract for the fact that the DPAA is of finite size by using a window function," Page 12 Line 25 – Page 13 Line 6 and Figures 11A – 11D, "The window function reduces the effects of "side lobes" at the expense of power. The type of window function used is chosen dependent on the qualities required of the resultant beam. Thus, if beam directivity is important, a window function as is shown in Figure 11A should be used. If less directivity is important, a window function as is shown in Figure 11D can be used," Page 26 Lines 20 – 25, "Thus, in general, output signals destined for SETs near the centre of the array will not be significantly affected but those near to the perimeter of the array will be attenuated according to how near to the edge of the array they are," Page 12 Line 25 – Page 13 Line 6.

Again, while not explicitly stated by Bienek, North discloses a similar speaker system with multiple transducers positioned in a plane for optimum acoustic radiation pattern and illustrates in Figures 20 – 23 that the frequency response to the audio signal at a second point in the sound field (off-axis) is lower than the frequency response to the audio signal at the first point in the sound field (on-axis). North explains "conventional arrays direct a portion of their sound towards the listener (main lobe) and a portion of their sound towards the sides (side lobes). When placed in a room, the sound waves in the side lobes reflect from surfaces such as walls, the ceiling, and the floor (room reflections). The reflected sound waves interact either constructively or destructively, depending on the delay time and frequency, with the direct sound waves. The listener is presented with a sound field that is a combination of the direct sound from the main lobe and the reflected and delayed sound from the side lobes. Although the speaker system may have exhibited a flat frequency response in an anechoic chamber, the frequency response in the room at the listener is anything but flat, with pronounced variations in the response. The result is the listener hears a severely distorted sound field that bears little resemblance to the original event," Column 2 Lines 22 – 52. North further illustrates on-axis and off-axis frequency response "waterfall" plots for different speaker arrays (Figures 20 – 23). North explains, "A waterfall plot illustrates the sound dispersion characteristics of a speaker system by cascading the multiple frequency response curves of a speaker system taken at multiple angles. The waterfall plots of FIGS. 20 through 23 represent 10 frequency response measurements taken in 10-degree increments from 0 degrees (on-axis) to 90 degrees off-axis. The

measurement curves cascade diagonally on top of one another, starting with the 0-degree on-axis curve (the rearmost curve) and ending with the 90-degree off-axis curve (the curve at the front), Column 6 Lines 34 – 44. North further explains, "with less energy directed towards the room walls, floor, and ceiling, the intensity of the room reflections decreases. This decreases the level of sounds that echoes within the room. A decreased level of room reflections can also result in a flatter frequency response in a room as the attenuated reflected waves interfere less with the sound waves in the main lobe," Column 7 Lines 32 – 45.

Thus in all cases, North illustrates (Figures 20 – 23) that the frequency response to the audio signal at a second point in the sound field (off-axis) is lower than the frequency response to the audio signal at the first point in the sound field (on-axis), especially Figure 22, where North illustrates that a significant reduction to off-axis sound energy due to increased directivity results "in a flatter frequency response in a room as the attenuated reflected waves interfere less with the sound waves in the main lobe," Column 7 Lines 32 – 45.

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention given the teachings of North to utilize the reduction in the effects of "side lobes" by using the amplitude control means by applying a window function to the audio signal as disclosed by Bienek to achieve a frequency response to the audio signal at a second point in the sound field (off-axis) that is lower than the frequency response to the audio signal at the first point in the sound field (on-axis) since it is beneficial to have a significant reduction to off-axis sound energy due to increased directivity given it

results "in a flatter frequency response in a room as the attenuated reflected waves interfere less with the sound waves in the main lobe," Column 7 Lines 32 – 45.

Bienek and North further do not disclose passing the audio signal through a variable high pass filter that is adjusted to filter the audio signal such that the frequency response to the audio signal at the second point in the sound field is suppressed relative to the frequency response to the audio signal at the first point in the sound field. However, in line with the teachings of Bienek and North of "reducing the effects of "side lobes"," Bienek Page 26 Lines 20 – 25, and having "a flatter frequency response in a room as the attenuated reflected waves interfere less with the sound waves in the main lobe," North Column 7 Lines 32 – 45, Kunugi teaches another method that does more than just reduce "side lobes" it eliminates the unintended sound waves at the listening point in the sound field while compensating for sound waves of insufficient low frequency level. Kunugi explains, "As is apparent from the above description, in the sound field correcting system according to this embodiment of the invention, a signal obtained by varying the frequency characteristic of the original signal so that the reflected sound waves are eliminated at the listening point is added to the original signal to obtain the loudspeaker driving signal. Therefore, the frequency characteristic at the listening point can be made essentially flat. Furthermore, the system is simple both in the adjustment required and in its construction, and it can be manufactured at a relatively low cost," Kunugi Column 7 Lines 35 – 45. Kunugi further improves upon the first embodiment and explains, "As is apparent from the above description, in accordance with the second embodiment of the invention, the reflected sound wave is

cancelled by controlling the delay and the level of the original signal, and furthermore the HPF having the cutoff frequency f_0 is inserted in the original signals's delay and level controlling system to boost the low frequency level. In this embodiment, the cutoff frequency f_0 of the HPF 2-9 is determined according to the sound field characteristic established by the direct sound L.sub.1 and the reflected sound path L_2. However, in practice, the HPF 2-9 should be implemented with a variable resistor R_1, as shown in FIG. 11, so that the cutoff frequency f_0 can be selected as required. In FIG. 11, C_1 designates a capacitor. According to the second embodiment of the invention, the HPF is used to change the frequency characteristic of the signal which is added to the original signal, and therefore sound waves at the listening point are remarkably improved in low frequency level. That is, sound waves of insufficient low frequency level are compensated for, with the result that the low frequency reproduction capability is substantially increased (Column 8 Lines 33 – 58).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the circuit of Bienek and North, to eliminate the unintended sound components responsible for excess reverberation, poor channel separation, and irregular frequency characteristics as taught by Kunugi (Column 3 Lines 16 – 31) and further include a variable high pass filter as taught by Kunugi, allowing for the low frequency response level of the audio signal at the listening point to be boosted relative to the reflected waves at the second point which are suppressed through cancellation, thus improving the frequency response at the listening point of Bienek and North.

Claim 11: Bienek, North, and Kunugi disclose the audio signal processor according to claim 10, wherein a sound wave from the speaker array is caused to reach at least one of the first and second points after it is reflected by a wall surface (Bienek Figure 8).

Claim 12: Bienek, North, and Kunugi disclose the audio signal processor according to claim 10, wherein when forming the first and second points in the sound filter, a filter factor of each of the plurality of digital filters is determined by calculation and set for each of the plurality of digital filters (Bienek Third Sound Field, Pages 21 – 22 and Figure 7C).

Claim 13: Bienek, North, and Kunugi disclose the audio signal processor according to claim 10, wherein when forming the first and second points in the sound field, a filter factor of each of the plurality of digital filters is read from a data base and set for each of the plurality of digital filters (stored sets of delays (for the DDGs) and filter coefficients (for the ADFS) can be recalled, Bienek Page 14 Lines 26 and 27).

6. Claims 5 – 9 and 14 – 20 are rejected under 35 U.S.C. 103(a) as being unpatentable over Bienek, North, and Kunugi in view of Masako et al. (JP-8-191225-A), hereinafter Masako.

Claim 5: Bienek, North, and Kunugi disclose the audio signal processing method according to claim 1, but does not explicitly disclose wherein: the predetermined delay time set for at least one of the plurality of digital filters is divided into an integer part and decimal part in units of a sampling period of the audio signal; over-sampling an impulse response including a delay time represented by at least the decimal part of the predetermined delay time for a shorter period than a sampling period to provide a sample train and, wherein the sample train is down-sampled to provide pulse-waveform data of the sampling period; and factor data is set for a part to be delayed by the plurality of digital filters based on the pulse- waveform data. Bienek does disclose "the minimum delay possible for a given signal is preferably as small or smaller than Ts, that signal's sample period" and that "most preferably, the smallest incremental change in delay possible for a given digital signal should be no larger than Ts, that signal's sampling period. Otherwise, interpolation of the signal is necessary," Page 12 Lines 17 – 24. Therefore Bienek does disclose a fractional delay and also discloses that a delay filter and an adaptive digital filter may be used. Bienek does not disclose details of how to perform, for example, the interpolation necessary for the fractional delays disclosed above and therefore one would be inclined to look elsewhere for such a teaching. Masako discloses the technique necessary to include a fractional delay (Figure 6 –7 and Paragraph 29 – 33) and discloses that this technique is an effective approach when slight spacing differences influence the felling of the direction of perceived sound. Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to include the process of over-sampling an impulse response and then down-

sampling to provide a pulse-waveform data of the sampling period as disclosed by Masako for setting the delay coefficients for the plurality of digital filters as disclosed by Bienek, North, and Kunugi, thereby providing better perceived spatial resolution.

Claim 6: Bienek, North, Kunugi, and Masako disclose the audio signal processing method according to claim 5, wherein the audio signal is delayed by a part of the predetermined delay time, which is a multiple of the sampling period, by digital delay circuits which operate for the sampling period, while it is being delayed by the remainder of the predetermined delay time, which includes the decimal part by the digital filters (Bienek discloses that the delay time may be a fractional sampling period and also discloses cascading a delay means with adjustable digital filter means that can also apply delays. Therefore given the disclosure of Bienek and the teachings of Masako of how to calculate a finer representation of an impulse response, Bienek, North, and Masako disclose implementing a delay using a simple delay element and an adjustable digital filter for the remainder or fractional part of the delay in a two stage process as disclosed by Bienek).

Claim 7: Bienek, North, Kunugi, and Masako disclose the audio signal processing method according to claim 5, and wherein: an over-sampling period of the over-sampling operation is 1/N (N is an integer larger than or equal to 2) of the sampling period of the digital signal; and when the delay time represented by the decimal part is

nearly an integral multiple (m) of the over-sampling period, m/N is adopted as the decimal part (Masako, Paragraph 26).

Claim 8: Bienek, North, Kunugi, and Masako disclose the audio signal processing method according to claim 7, wherein: the pulse-waveform data to be delayed by a delay time which is m/N (m = 1 to N - 1) of the sampling period is pre-stored in a data base; and pulse-waveform data approximate to the decimal part is taken out of the stored pulse-waveform data and set as a filter factor of each of the plurality of digital filters (Bienek, stored sets of delays (for the DDGs) and filter coefficients (for the ADFS) can be recalled, Page 14 Lines 26 and 27).

Claim 9: Bienek, North, Kunugi, and Masako disclose the audio signal processing method according to claim 5, wherein a transfer characteristic providing a predetermined acoustic effect is convoluted in the pulse-waveform data and set as a filter factor of each of the plurality of digital filters ("convolution multiplier", Masako, Paragraph 26).

Claim 14: Bienek, North, and Kunugi disclose the audio signal processor according to claim 10, wherein: the pulse-waveform provided by the calculation circuit is set as a filter factor of each of the plurality of digital filters (Third Sound Field, Pages 21 – 22 and Figure 7C) but does not explicitly disclose the predetermined delay time set for at least one of the plurality of digital filters is divided into an integer part and decimal part in

units of a sampling period of the audio signal, there is further provided a calculation circuit to calculate pulse-waveform data of the sampling period by over-sampling an impulse response including a delay time represented by at least the decimal part of the predetermined delay time for a shorter period than the sampling period to provide a sample train, and down-sampling the sample train. Bienek does disclose "the minimum delay possible for a given signal is preferably as small or smaller than Ts, that signal's sample period" and that "most preferably, the smallest incremental change in delay possible for a given digital signal should be no larger than Ts, that signal's sampling period. Otherwise, interpolation of the signal is necessary," Page 12 Lines 17 – 24. Therefore Bienek does disclose a fractional delay and also discloses that a delay filter and an adaptive digital filter may be used. Bienek does not disclose details of how to perform, for example, the interpolation necessary for the fractional delays disclosed above and therefore one would be inclined to look elsewhere for such a teaching. Masako discloses the technique necessary to include a fractional delay (Figure 6 –7 and Paragraph 29 – 33) and discloses that this technique is an effective approach when slight spacing differences influence the felling of the direction of perceived sound. Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to include the process of over-sampling an impulse response and then down-sampling to provide a pulse-waveform data of the sampling period as disclosed by Masako for setting the delay coefficients for the plurality of digital filters as disclosed by Bienek, North, and Kunugi, thereby providing better perceived spatial resolution.

Claim 15: Bienek, North, Kunugi, and Masako disclose the audio signal processor according to claim 14, wherein: an over-sampling period of the over-sampling in the calculation circuit is 1/N (N is an integer larger than or equal to 2) of the sampling period of the digital signal; and when the delay time represented by the decimal part is nearly an integral multiple (m) of the over-sampling period, m/N is adopted as the decimal part (Masako, Paragraph 26).

Claim 16: Bienek, North, Kunugi, and Masako disclose the audio signal processor according to claim 14, wherein a transfer characteristic providing a predetermined acoustic effect is convoluted in the pulse-waveform data to set synthetic-waveform data as a filter factor of each of the plurality of digital filters ("convolution multiplier", Masako, Paragraph 26).

Claim 17: Bienek, North, and Kunugi disclose the audio signal processor according to claim 10, wherein: the pulse-waveform data stored in the storing means is taken out and set as a filter factor of each of the plurality of digital filters (stored sets of delays (for the DDGs) and filter coefficients (for the ADFS) can be recalled, Page 14 Lines 26 and 27) but does not explicitly disclose the predetermined delay time set for at least one of the plurality of digital filters is divided into an integer part and decimal part in units of a sampling period of the audio Signal; there is further provided a storing means for storing pulse-waveform data of the sampling period provided by over- sampling an impulse response including a delay time represented by at least the decimal part of the

predetermined delay time for a shorter period than the sampling period to provide a sample train, and down-sampling the sample train. Bienek does disclose "the minimum delay possible for a given signal is preferably as small or smaller than Ts, that signal's sample period" and that "most preferably, the smallest incremental change in delay possible for a given digital signal should be no larger than Ts, that signal's sampling period. Otherwise, interpolation of the signal is necessary," Page 12 Lines 17 – 24. Therefore Bienek does disclose a fractional delay and also discloses that a delay filter and an adaptive digital filter may be used. Bienek does not disclose details of how to perform, for example, the interpolation necessary for the fractional delays disclosed above and therefore one would be inclined to look elsewhere for such a teaching. Masako discloses the technique necessary to include a fractional delay (Figure 6 –7 and Paragraph 29 – 33) and discloses that this technique is an effective approach when slight spacing differences influence the felling of the direction of perceived sound. Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to include the process of over-sampling an impulse response and then down-sampling to provide a pulse-waveform data of the sampling period as disclosed by Masako for setting the delay coefficients for the plurality of digital filters as disclosed by Bienek, North, and Kunugi, thereby providing better perceived spatial resolution.

Claim 18: Bienek, North, Kunugi, and Masako disclose the audio signal processor according to claim 17, wherein: an over-sampling period of the over-sampling is 1/N (N is an integer larger than or equal to 2) of the sampling period of the digital signal; and

when the delay time represented by the decimal part is nearly an integral multiple (m) of the over-sampling period, m/N is adopted as the decimal part (Masako, Paragraph 26).

Claim 19: Bienek, North, Kunugi, and Masako disclose the audio signal processor according to claim 17, wherein: a plurality of the pulse-waveform data corresponding to the decimal part is pre-stored in the storing means; and pulse-waveform data approximate to the decimal part is taken out of the stored pulse-waveform data and set as a filter factor of each of the plurality of digital filters (Bienek, stored sets of delays (for the DDGs) and filter coefficients (for the ADFS) can be recalled, Page 14 Lines 26 and 27).

Claim 20: Bienek, North, Kunugi, and Masako disclose the audio signal processor according to claim 17, wherein a transfer characteristic providing a predetermined acoustic effect is convoluted in the pulse-waveform data to set the pulse-waveform data as a filter factor of each of the plurality of digital filters ("convolution multiplier", Masako, Paragraph 26).

Response to Arguments

7. Applicant's arguments with respect to claims 1 – 20 have been considered but are moot in view of the new ground(s) of rejection in view of Kunugi et al. (US 4,868,878).

Conclusion

8. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Joseph Saunders whose telephone number is (571) 270-1063. The examiner can normally be reached on Monday - Thursday, 9:00 a.m. - 4:00 p.m., EST.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Vivian Chin can be reached on (571) 272-7848. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

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/J. S./
Examiner, Art Unit 2614

/Vivian Chin/
Supervisory Patent Examiner, Art Unit 2614